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Robin E. Barnum

In re Application of

Stephen S. Oh, et al.

Serial No: 09/483,569

Filed: January 14, 2000

For: Simplified Noise Suppression Circuit

Conf. No: 8551

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Sir:

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of		TI-23373
	Stephen S. Oh, et al.	Art Unit: 2654
Serial No.:	09/483,569	Examiner: Michael N. Opsasnick
Filed:	January 14, 2000	Conf. No.: 8551
For:	Simplified Noise Suppression Circuit	

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___ DECLARATION (# Pages)	___ NOTICE OF APPEAL (Page)
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___ CONTINUATION APP'N (# Pages)	X Appeal Brief TL (1 page)
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NAME OF INVENTOR(S): Stephen S. Oh, et al.	RECEIPT DATE & SERIAL NO.: **Serial No.: 09/483,569** **Filing Date:** January 14, 2000
TITLE OF INVENTION: Simplified Noise Suppression Circuit	
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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant: Oh et al
Serial No.: 09/483,569
Filed: January 14, 2000
For: SIMPLIFIED NOISE SUPPRESSION CIRCUIT

Art Unit: 2654
Examiner: Michael N. Opsasnick
Docket: TI-23373

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Appeal Brief under 37 C.F.R. §41.37

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Rob E. Barn

Robin E. Barnum

Dear Sir:

This is Appellant's Appeal Brief filed pursuant to 37 C.F.R. §41.37 and the Notice of Appeal filed July 28, 2005.

TABLE OF CONTENTS

Real Party in Interest

The real party in interest in this application is Texas Instruments Incorporated, a corporation of Delaware with its principal place of business in Dallas, Texas. An assignment to Texas Instruments Incorporated is recorded at reel 010513 and frames 0488 to 0490.

Related Appeals and Interferences

There are no appeals of interferences related to this appeal in this application.

Status of the Claims

Claims 1 to 3 and 9 to 11 are finally rejected. Claims 4 to 8 and 12 to 22 are canceled. No claims are allowed.

Status of Amendments Filed After Final Rejection

No amendments to the claims were proposed in the response filed April 5, 2005 following the FINAL REJECTION of February 7, 2005.

Summary of Claimed Subject Matter

This invention is a method and apparatus for reducing noise in a sampled acoustic signal. A sampler obtains discrete samples of an acoustic signal. An analog to digital converter forms a stream of sampled acoustic signals. The invention selects a fixed number of samples. This fixed number of samples is preferably 32 samples. The invention multiplies these samples by a windowing function. This windowing function is preferably a hanning window function. A fast Fourier transform of the windowed samples yields transformed windowed signals. The invention selects half of the transformed windowed signals. The invention calculates a power estimate of the transformed windowed signals and a smoothed power estimate by

smoothing the power estimate over time. The invention calculates a noise estimate. Then invention calculates a gain function from the noise estimate and the smoothed power estimate. The invention calculates a transformed speech signal by multiplying the gain function with the transformed windowed signal. An inversed fast Fourier transform of the transformed speech signal yields a sampled speech signal. The invention adds the sampled speech signal to a portion of the speech signal of a previous frame.

Grounds for Rejection to be Reviewed on Appeal

Claims 1 to 3 and 9 to 11 were rejected under 35 U.S.C. 103(a) as made obvious by Bloebaum et al, U.S. Patent No. 6,070,137.

Arguments

Claims 1 and 9 recite subject matter not made obvious by Bloebaum et al. Claim 1 recites "calculating a smoothed power estimate by smoothing the power estimate over time." Claim 9 recites the noise suppression circuit operates to "calculate a smoothed power estimate by smoothing the power estimate over time." The FINAL REJECTION demonstrates that Bloebaum et al fails to make this limitation obvious. In particular, the FINAL REJECTION shows that Bloebaum et al teaches smoothing over time of a different signal than that claimed in claims 1 and 9.

Claim 1 recites calculation of "a gain function from the noise estimate and the smoothed power estimate." Claim 9 recites the noise suppression circuit operates to "calculate a gain function from the noise estimate and the smoothed power estimate." The FINAL REJECTION states at page 4, lines 11 and 12 that Bloebaum et al:

> "• calculates a gain function from the signal and noise power estimates (enhancement filter, col. 6, lines 8-10);"

This portion of the FINAL REJECTION refers to Figure 4 of Bloebaum et al. This Figure 4 illustrates transform and filter computation block 56 receiving the power spectral density (PSD) estimate represented by $|S(e^{j\omega})|^2$ from block 44 and the noise vector N from noise model adaptation block 46 and producing enhancement filter $|H(e^{j\omega})|$. In order for the Examiner's statement at page 4, lines 11 to 12 of the FINAL REJECTION to be true, one input to transform and filter computation block 56 must correspond to the claimed noise estimate and the other input must correspond to the claimed smoothed power estimate. Bloebaum et al states at column 5, lines 58 and 59:

> "The forward transform G converts the noise vector N into the noise PSD estimate $|N(e^{j\omega})|^2$."

Thus this input to transform and filter computation block 56 must correspond to the claimed noise estimate. Accordingly, the other input to transform and filter computation block 56 $|S\hat{}(e^{j\omega})|^2$ must correspond to the claimed smoothed power estimate. However, Bloebaum et al fails to teach that this input $|S\hat{}(e^{j\omega})|^2$ is smoothed over time as required by the language of claims 1 and 9. Bloebaum et al states at column 5, lines 60 to 62 referring to variance reduction block 58:

> "The Variance Reduction block receives as input $|S(e^{j\omega})|^2$ and applies a smoothing function in the frequency domain to generate an output $|S\hat{}(e^{j\omega})|^2$."

Thus Bloebaum et al clearly teaches $|S\hat{}(e^{j\omega})|^2$ is smoothed in the frequency domain and not smoothed over time as recited in claims 1 and 9. The Applicant respectfully submits that disclosure of smoothing in the frequency domain fails to make obvious the smoothing over time of claims 1 and 9.

In summary, Bloebaum et al teaches a calculation of a gain or filter function in transform and filter computation block 56 similar to the recitations of claims 1 and 9. In Bloebaum et al, one input $|N(e^{j\omega})|^2$ is related to the noise estimate and the other input $|S\hat{}(e^{j\omega})|^2$ is related to the power estimate. Bloebaum et al teaches the noise estimate $|N(e^{j\omega})|^2$ is smoothed over time (equation at column 5, line 40) and the power estimate $|S\hat{}(e^{j\omega})|^2$ is smoothed over frequency (column 5, lines 60 to 62). Claims 1 and 9 recite smoothing the power estimate over time. Bloebaum et al thus teaches smoothing over time of a different signal than that recited in claims 1 and 9 and frequency smoothing of the signal that claims

1 and 9 recite as time smoothed. Accordingly, Bloebaum et al fails to make obvious claims 1 and 9.

The FINAL REJECTION states at page 4, lines 5 to 10 that Bloebaum et al teaches:

> "• calculating a smoothed power estimate over time by smoothing the power estimate using the recited (i.e., first-order AR smoothing) equation (Fig. 5, element 64 with 'smoothed version of S' in col. 8, lines 6-8; *cf.* first order AR smoothing, col. 5, lines 38-44), wherein
> "• noting that S is signal power with signal present and noise power when signal absent, thus also calculating a noise estimate"

The Applicants submit that the signal N supplied to transform and filter computation block 56 from noise model adaption block 46 is only a noise estimate and includes no signal. Bloebaum et al states at column 5, lines 21 to 45:

> "An important aspect of integrating noise suppression into the MBE speech encoder 20 is the computation of a model of the background noise. The noise model in FIG. 3 is represented as a vector N output from a noise model adaptation block 46. This invention is not restricted to any particular method of modeling background noise, and several possible methods are discussed herein. The noise model is stored by the noise model adaptation block 46 and is updated when the vadFlag is set to zero, indicating that there is an absence of speech. The adaptation process involves smoothing of the model parameters in order to reduce the variance of the noise estimate. This may be done using either a moving average (MA), autoregressive (AR), or a combination ARMA process. AR smoothing is the preferred technique, since it provides good smoothing for a low ordered filter. This reduces the memory storage requirements for the noise suppression algorithm. The noise model adaptation with first order AR smoothing is given by the following equation:
>
> $$N^{(i)} = \alpha N^{(i-1)} + (1-\alpha)S,$$
>
> where α may be in the range $0.1 \le \alpha \le 1$, but is further constrained to the range $0.8 \le \alpha \le 0.95$ in the preferred

embodiment of the invention. The vector S is an input to block 46 from a Transform and Filter Computation block 56."

The text of Bloebaum et al makes clear that the vector N is a noise model "output from a noise model adaptation block 46." The first order AR smoothing of the equation is used in adapting the noise model. This portion of Bloebaum et al teaches that the noise model "is updated when the vadFlag is set to zero, indicating that there is an absence of speech." Accordingly, the AR smoothing equation is employed only in the absence of signal in S and is employed only to update a "noise model is stored by the noise model adaptation block 46." This portion of Bloebaum et al clearly teaches smoothing of the vector N from noise model adaption block 46 as a function of the prior noise vector N and the vector S in the absence of signal. Thus this is not smoothing the power estimate as claimed. Claims 1 and 9 recite such a noise estimate as a different signal employed in the calculation of the gain function. Thus this equation fails to make obvious calculating "a smoothed power estimate by smoothing the power estimate over time" as recited in claims 1 and 9.

The FINAL REJECTION cites variance reduction 64 described in Bloebaum et al at column 8, lines 6 to 8 and illustrated in Figure 5 as teaching the recited smoothing over time with reference to Bloebaum et al at column 5, lines 38 to 44. Bloebaum et al at column 5, lines 38 to 44 teaches smoothing over time of the noise vector N produced by noise model adaptation block 46. This smoothing over time is not applicable to variance reduction 64 of Figure 5. Bloebaum et al states at column 8, lines 1 to 10:

"This alternate version is denoted by block 62 and is shown in FIG. 5. The principal novelty of the block 62 versus the block 56 is that the enhancement filter is computed in the domain of the noise model and then transformed to the sampled frequency domain. In FIG. 5, the signal model vector S is input to the

> Variance Reduction block 64, which outputs a smoothed version of S denoted S^. This vector S and the noise model vector N are input to the Enhancement Filter Computation block 66."

This teaching of Bloebaum et al fails to state that variance reduction block 64 smoothes over time as required by the language of claims 1 and 9. Because Figure 5 is taught as an alternative to Figure 4, one skilled in the art would believe that variance reduction block 64 operates similarly to analogous variance reduction block 58 of Figure 4. As quoted above, Bloebaum et al states at column 5, lines 60 to 62 variance reduction block 58 smoothes in the frequency domain. Accordingly, one skilled in the art would believe that variance reduction block 64 also smoothes in the frequency domain. Thus claims 1 and 9 are not made obvious by Bloebaum et al.

The FINAL REJECTION states at page 2, line 17 to page 3, line 3:

> "As for the assertion that Bloebaum's smoothed signal and power estimates are not used to compute his gain function (Amendment, p.9), this is clearly false, since the gain is computed in the special enhancement filter (element 52 of Figure 3), which clearly gets inputs from the smoothed signal power computation element (56) therein and the smoothed noise power computation element (46, its output going through element 56 on the way to element 52, as indicated by the corresponding arrows in Figure 1)."

The Applicants believe this statement mischaracterizes the Applicant's argument. The response filed October 29, 2004 states at page 8, line 29 to page 9, line 11:

> "Claims 1 and 9 each recite calculation of "a gain function from the noise estimate and the smoothed power estimate." Bloebaum et al illustrates transform and filter computation block 56 which receives the power spectral density (PSD) estimate represented by $|S(e^{j\omega})|^2$ from block 44 and the vector N from noise model adaption block 46 and produces

> enhancement filter $|H(e^{j\omega})|$. If the vector N is the claimed smoothed power estimate, then transform and filter computation block 56 receives the power spectral density estimate from block 44 and the smoothed power spectral density estimate (vector N) from noise model adaption block 46. These are not the inputs to the calculated gain function recited in claims 1 and 9. Thus if the vector N is the claimed smoothed power estimate, Bloebaum et al fails to make obvious a different limitation of claims 1 and 9. Accordingly, the Appellants respectfully submit that claims 1 and 9 are allowable over Bloebaum et al."

This argument is conditional by the phrase "If the vector N is the claimed smoothed power estimate." Figure 3 of Bloebaum et al shows that transform and filter computation 56 receives power spectral density (PSD) estimate represented by $|S(e^{j\omega})|^2$ from block 44 and the vector $\underline{N}$ from noise model adaption block 46 and produces enhancement filter $|H(e^{j\omega})|$. The Applicants believe that power spectral density (PSD) $|S(e^{j\omega})|^2$ corresponds to the claimed power estimate and the vector $\underline{N}$ corresponds to the claimed noise estimate. In order for the time smoothing equation at Bloebaum et al column 5, line 40 to apply to the claimed power estimate, the Examiner must argue that the vector $\underline{N}$ from noise adaption block 46 is that power estimate. The above quoted portion of the response filed October 29, 2004 points out that this argument results in transform and filter computation block 56 receiving differing inputs than recited in the paragraphs of claims 1 and 9 which calculate a gain function. The Applicants do not believe that the conditional "If the vector N is the claimed smoothed power estimate" is true. The Applicants submit that if this conditional is true, the Examiner's rejection fails relative to another limitation of claims 1 and 9. Thus this argument points out an inconsistency in the Examiner's rejection.

Claims 2, 3, 10 and 11 are allowable by dependency upon allowable base claims.

If the Examiner has any questions or other correspondence regarding this application, Applicants request that the Examiner contact Applicants' attorney at the below listed telephone number and address to facilitate prosecution.

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APPENDIX

CLAIMS ON APPEAL

1. (Previously Presented) A method for reducing noise in a sampled acoustic signal, comprising:

receiving a stream of sampled acoustic signals;

digitizing each sampled acoustic signal thereby forming digital samples;

selecting a fixed number of digital samples;

multiplying the digital samples by a windowing function;

computing the fast Fourier transform of the selected windowed digital samples to yield transformed windowed signals;

selecting half of the transformed windowed signals;

calculating a power estimate of the transformed windowed signals;

calculating a smoothed power estimate by smoothing the power estimate over time using the equation:

$$P^{t}(i) = (1-\alpha)\ P^{t-1}(i) + \alpha\ P(i)$$

where: $P^{t}(i)$ is the smoothed power estimate for a current time sample to be calculated for the i-th FFT point; $P^{t-1}(i)$ is the smoothed power estimate for an immediately prior time sample for the i-th FFT point; $P(i)$ is the calculated power estimate of the transformed windowed signals for the i-th FFT point; and α is an experimentally chosen predetermined value called the smoothing factor;

calculating a noise estimate;

calculating a gain function from the noise estimate and the smoothed power estimate;

calculating a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculating an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

adding the sampled speech signal to a portion of the speech signal of a previous frame.

2. (Original) The method of Claim 1, wherein the fixed number of samples is thirty-two.

3. (Original) The method of Claim 1, wherein the windowing function is a hanning window function.

9. (Previously Presented) A system for reducing noise in an acoustical signal comprising:

a sampler for obtaining discrete samples of the acoustical signal;

an analog to digital converter coupled to the sampler an operable to convert the analog discrete samples into a digitized sample;

a noise suppression circuit coupled to the analog to digital converter and operable to:

receive the digitized samples;

select a fixed number of digitized samples;

multiply the digitized samples by a windowing function;

compute the fast Fourier transform of the windowed digitized samples to yield transformed windowed signals;

select half of the transformed windowed signals;

calculate a power estimate of the transformed windowed signals;

calculate a smoothed power estimate by smoothing the power estimate over time using the equation:

$$P^{t}(i) = (1-\alpha)\ P^{t-1}(i) + \alpha\ P(i)$$

where: $P^t(i)$ is the smoothed power estimate for a current time sample to be calculated for the i-th FFT point; $P^{t-1}(i)$ is the smoothed power estimate for an immediately prior time sample for the i-th FFT point; $P(i)$ is the calculated power estimate of the transformed windowed signals for the i-th FFT point; and α is an experimentally chosen predetermined value called the smoothing factor;

calculate a noise estimate;

calculate a gain function from the noise estimate and the smoothed power estimate;

calculate a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculate an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

add the sampled speech signal to a portion of the speech signal of a previous frame.

10. (Original) The system of Claim 9, wherein the fixed number of samples is thirty-two.

11. (Original) The system of Claim 9, wherein the windowing function is a hanning window function.